Adoul discloses a method of searching a codebook in association with the process of decoding a sound signal. Candidate search paths are selected by means of a likelihood-estimate vector based on speech-related signals

Osawa discloses a method for speech coding. According to Osawa, an excitation quantizer searches for the best set of positions of a plurality of pulses representing a speech signal within an area determined in reference to positions selected as those meeting with preassigned conditions for a pitch prediction signal obtained accordingly to an adaptive codebook. According to this method, it becomes possible to achieve a higher quality of sounds than possible with a prior art method under a situation in which the associated bit rate is being reduced, because the area being searched for pulse positions is determined in a way suited to a relevant pitch waveform so that the pulse positions can be selected to describe in a good quality the speech signal associated with the pitch waveform. It is configured further to comprise determining a plurality of modes by extracting feature amounts from an inputted speech so that the excitation quantizer can perform the above processes in a preassigned mode, and consequently achieving a high quality level of sound for those of the highly periodic mode.

Applicants' claimed invention (for example, as claimed in independent claims 1, 7, 13 and 16) discloses a speech coding and decoding method and apparatus that involves the vector-quantization of the analysis-by-synthesis type. It, in contrast to the above two cited references, uses a configuration variable codebook. Each of the code-words contained in it is constituted only from a plurality of non-zero amplitude values. Sample positions associated with the non-zero amplitude values are variably controlled using

both an index i and a transmission parameter ρ which represents the feature amount of a voice. The transmission parameter may be constituted by a lag value corresponding to a pitch period, or alternatively, by a pitch gain value. It is also possible to configure the voice coding method in such a manner that the sample positions of non-zero amplitude values are revised within the region corresponding to the lag value in association with the comparative relation between lag value magnitudes or with the pitch gain value. FIGs. 5 and 6 respectively illustrate coding and decoding embodiments of the present invention. In the drawings. 1 and 1' indicate a configuration variable codebook, 2 and 2' indicate a gain unit, 3 and 3' indicate a linear prediction synthesis filter, 4 indicates a subtractor and 5 indicates an error power evaluation unit. The configuration variable codebook 1 or 1' may be considered, for example, an algebraic codebook for outputting code vectors each constituted from a plurality of non-zero sample values. It is capable of revising its structure by adjusting non-zero sample positions with reference to the values of index i and the transmission parameter ρ such as a value representing the pitch period (lag value). Importantly, in adjusting non-zero sample positions in a standard frame, for example, for lower bit-rate voice signals, according to Applicants' inventive method and in sharp contrast to prior algebraic codebook-based methods, the configuration variable codebook 1 or 1' does not change the number of the involved non-zero samples. According to this structure of the method, it becomes possible to better control the increase in number of the transmission bits required for indexing code-words.

These claimed features are not discloses or suggested by Adoul or Ozawa. Accordingly, Applicants respectfully submit that claims 1, 7, 13 and 16 stand in condition for allowance.

CONCLUSION

An earnest effort has been made to be fully responsive to the Examiner's objections. In view of the above amendments and remarks, it is believed that claims 1 - 18, consisting of independent claims 1, 7, 13 and 16, and the claims dependent therefrom, are in condition for allowance. Passage of this case to allowance is earnestly solicited. However, if for any reason the Examiner should consider this application not to be in condition for allowance, he is respectfully requested to telephone the undersigned attorney at the number listed below prior to issuing a further Action.

Attached is a marked up version of the changes made to the specification and claims by the current amendment. The attached pages are captioned **"Version With Markings To Show Changes Made"**.

Any fee due with this paper may be charged on Deposit Account 50-1290.

Respectfully submitted,

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Version with Markings to Show Changes Made – S/N 09/386,824

IN THE CLAIMS

1. (Amended) A voice coding method based on analysis-by-synthesis vector quantization [using a code book containing a voice source code vector having only a plurality of non-zero amplitude values,] comprising [the step of]:

using a configuration variable code book containing a voice source code vector having only a plurality of non-zero amplitude values; and

variably controlling a position of a sample of the non-zero amplitude value in the configuration variable code book using an index and a transmission parameter indicating a feature amount of voice.

2. (Amended) The method according to claim 1, further comprising [the step of]:

variably controlling the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

3. (Amended) The method according to claim 2, further comprising [the step of]:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value

depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

4. **(Amended)** The method according to claim 1, further comprising [the step of]<u>:</u>

variably controlling the position of the sample of the non-zero amplitude value <u>in the configuration variable code book</u> using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

5. **(Amended)** The method according to claim 4, further comprising [the step of]<u>:</u>

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

6. **(Amended)** The method according to claim 5, further comprising [the step of]<u>:</u>

reconstructing the position of the sample the non-zero amplitude value <u>in the configuration variable code book</u> within a region corresponding to the lag value depending on the pitch gain value.

7. **(Amended)** A voice decoding method for decoding a voice signal coded by a voice coding method based on analysis-by-synthesis vector quantization [using a code book containing a voice source code vector having only a plurality of non-zero amplitude values,] comprising [the step of]:

using a configuration variable code book containing a voice source code vector having only a plurality of non-zero amplitude values; and

variably controlling a position of a sample of the non- zero amplitude value in the configuration variable code book using an index and a transmission parameter indicating a feature amount of voice.

8. **(Amended)** The method according to claim 7, further comprising [the step of]:

variably controlling the position of the sample of the non-zero amplitude value in the configuration variable code book using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

9. **(Amended)** The method according to claim 8, further comprising [the step of]:

reconstructing the position of the sample of the non-zero amplitude value in the configuration variable code book within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a ceding unit of the voice.

10. (Amended) The method according to claim 7, further comprising [the step of]<u>:</u>

variably controlling the position of the sample of the non-zero amplitude value <u>in the configuration variable code book</u> using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

11. (Amended) The method according to claim 10, further comprising [the step of]<u>:</u>

reconstructing the position of the sample of the non-zero amplitude value <u>in the configuration variable code book</u> within a region corresponding to the lag value depending on a relationship between the lag value and a frame length which is a coding unit of the voice.

12. (Amended) The method according to claim 11, further comprising [the step of]<u>:</u>

reconstructing the position of the sample of the non-zero amplitude value <u>in the configuration variable code book</u> within a region corresponding to the lag value depending on the pitch gain value.

13. (Amended) A voice coding apparatus based on analysis-by- synthesis vector quantization [using a coda bock containing a voice source code vector having only a plurality of non-zero amplitude values,] comprising:

a configuration variable code book unit containing a voice source code vector having only a plurality non-zero amplitude values, wherein

[a] said configuration variable code book unit variably [controlling] controls a position of a sample of the non-zero amplitude value in said configuration variable code book unit using an index and a transmission parameter indicating a feature amount of voice.

14. (Amended) The apparatus according to claim 13, wherein:

said configuration variable code book unit variably controls the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

15. (Amended) The apparatus according to claim 13, wherein:

said configuration variable code book unit variably controls the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.

16. (Amended) A voice decoding apparatus for decoding a voice signal coded by a voice coding apparatus based on analysis-by-synthesis vector quantization [using a code book

containing a voice source code vector having only a plurality of non-zero amplitude values,] comprising:

a configuration variable code book unit containing a voice source code vector having only a plurality of non-zero amplitude values, wherein

[a] said configuration variable code book unit variably [controlling] controls a position of a sample of the non-zero amplitude value using an index and a transmission parameter indicating a feature amount of voice.

17. (Amended) The apparatus according to claim 16, wherein:

said configuration variable code book unit variably controls the position of the sample of the non-hero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice.

18. (Amended) The apparatus according to claim 16, wherein:

said configuration variable code book unit variably controls the position of the sample of the non-zero amplitude value in said configuration variable code book unit using the index and a lag value corresponding to a pitch period which is a transmission parameter indicating the feature amount of voice and a pitch gain value.